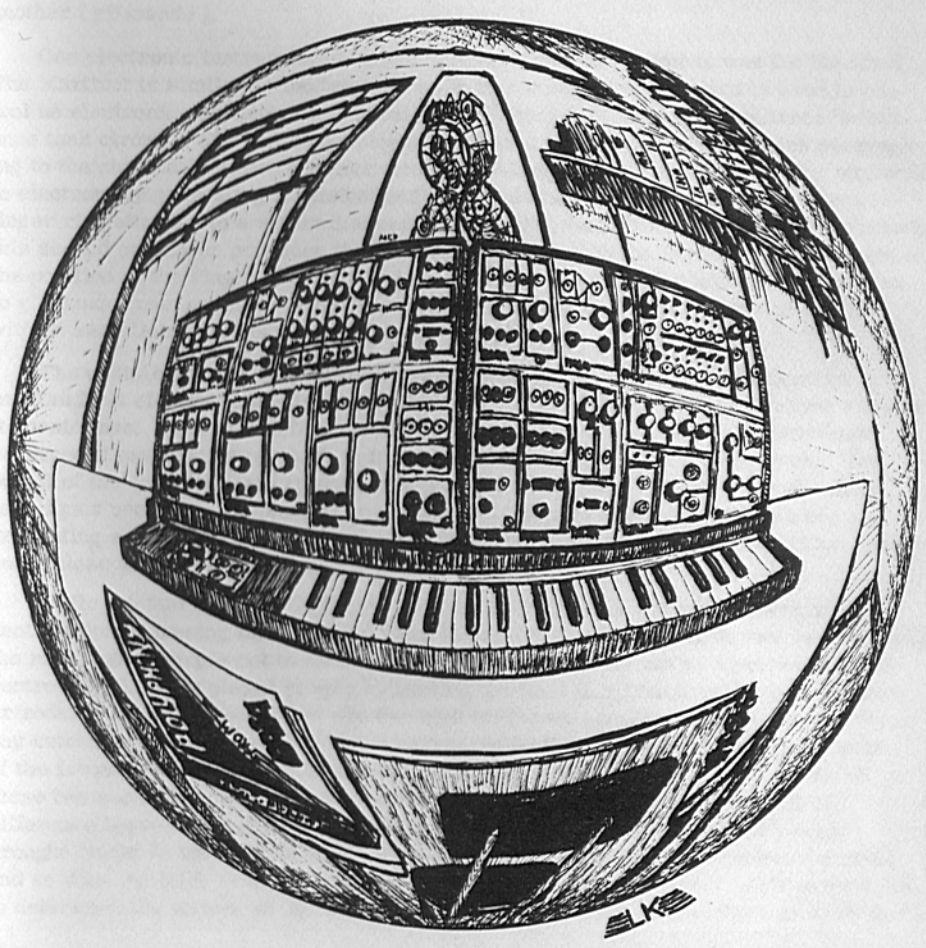
A GUIDE FOR USING THE PAIA 4700/S SYNTHESIZER SYSTEM



© 1976 PAIA Electronics, Inc. 1020 W. Wilshire Blvd., Okla. City, OK 73116

There has never been a musical instrument that was concieved overnight and released to the world in an immutable state the next day, each instrument has undergone change and refinement to bring it to its present condition. The same is true of electronic music but the newness of this field coupled with the technology explosion has caused its development to be compressed in time. Whereas the piano has taken centuries to evolve, electronic musical instruments first appeared only a little over four decades ago.

Most of these first instruments were little more than exercises in technology but some were designed to overcome shortcomings in existing instruments. For instance, the piano keyboard is one of the most powerful musical operating systems available but it has one outstanding drawback in that while it provides the musician easy access to all twelve notes of the equally tempered musical scale it prohibits him from using the infinity of musical pitches between those twelve notes. By its very nature it eliminates the possibility of any easy glide from one musical pitch to another (glissando).

One electronic instrument designed to overcome this weakness was the Martinot. The Martinot is similar to modern organs in that a standard keyboard is used to control an electronic oscillator built around a frequency determining capacitance/inductance tank circuit. The inductor is tapped at points that produce frequencies corresponding to the chromatic scale; a rather straightforward, if somewhat simplistic, approach to electronicly generating a musical scale. In addition to the keyboard there is a finger ring attached to a slider that controls another oscillator. When properly adjusted this second oscillator produces the pitch corresponding to the keyboard key adjacent to the position of the ring. The combination of keyboard and slider allows the musician to glissando from one note to another or add vibrato with a simple move of the hand without sacrificing the operating ease of the keyboard.

The ondioline was a contemporary of the Martinot but is significant because it was the first electronic musical instrument to use something other than a sine wave as its basic tone. In the Ondioline a relaxation oscillator controlled by the keyboard produced a sawtooth wave which in turn activated several frequency dividers. The output of the oscillator and frequency dividers were combined using much the same techniques used in some modern organs so that the instrument was capable of generating a great variety of sounds. Observers report that a skilled operator could come close to making the Ondioline talk.

While the Martinot and Ondioline were both designed in France, America's contribution to freeing the musician from the restrictions of the keyboard was probably the most outstanding - not to mention bizarre. A Theremin has no visible means of control at all and is played simply by moving the hand in relation to two metal plates or rods. Inside the instrument are two high frequency oscillators, one shielded from any external influences and the second arranged so that the plate or rod forms part of the frequency determining inductance/capacitance tank circuit. The outputs of these two oscillators are combined in such a way that an audible tone that is the difference between the two frequencies is produced. As the performer's hand is brought closer to the sensing antenna the difference in the two frequencies increases and so does the pitch of the tone. A second circuit allows the performer's other hand to determine the volume of the sound produced. Since there are no frets or keys to

to provide visual or tactile clues to the pitch a Theremin will produce, it is a very difficult instrument to play - but loads of fun.

SYNTHESIZERS

The first equipment that would come close to meeting our current definition of a synthesizer was built by Dr. Harry Olson during the early 1940's.* Produced under the auspices of the RCA Labs, the RCA Mark I and Mark II Synthesizers were something to behold. The Mark I has been disassembled for some time now but the Mark II still exists and is currently being leased to Columbia-Princeton Electronic Music Center, it neasures 17 feet long by 7 feet high and is valued at anywhere between $250,000 and one and a half million dollars depending on who you're talking to.

The average performer might be a little disappointed in the Mark II today because even if there were some way to transport it to a gig, he would find when he got there that he couldn't actually perform a number. The Mark II was simply not capable of real-time operation, each characteristic of the sound the instrument was to produce was laboriously calculated and plotted ahead of time and the result punched into a roll of paper tape. When it came time for the Mark II to do its thing the tape was fed in - like a very large, very expensive player piano - and the results recorded on a multi-track disc, (early 40's, remember, recording tape wasn't so hot in those days.) When all the parts of a number had been recorded on the separate tracks of the disc they were re-recorded on another disc from which a master was made.

You might think that about the only thing that the Mark I and Mark II did that was of any consequence was add the word Synthesizer to our vocabulary but that's not the case at all. They were significant first of all becuase they were the first to put it all together as far as electronic music production was concerned. All the oscillators, amplifiers and filters needed in one place at one time and best of all some means - no matter how cumbersome - of controlling them all. Secondly, they were the first instruments to utilize white noise sources as part of an electro-musical instrument. White noise will be covered in detail later, for now it should suffice to say that without it sounds like snare drums and cymbals, to mention only two, are impossible.

Don't get the impression that electronic music cannot be produced without a synthesizer, that's not true. Imagine that you are in a laboratory with all sorts of electronic equipment such as oscillators, filters, amplifiers, modulators, tape recorders, etc. You turn on one of the tape recorders and set the oscillator for the pitch you want, twiddle the knobs of the amplifier to shape the loudness contour and play with the filter knobs to adjust timbre. It only takes about six hands and a couple of minutes out when you're through you've got a whole note recorded on the tape. Repeat the process often enough and you've got a whole string of notes. Of course, the tempo is not right and the notes may not be in the right sequence but you can fix that by snipping the tape apart and editing out all the junk before splicing it back together again to produce the desired melody. Now you go back and do the same thing for bass, rhythm and all the other parts. About the only thing you can say for this technique is that it should certainly give you a feeling of accomplishment. Considering the complexity of the process even such monstrosities as dogs barking out the tune of "Away in a Manger" can be forgiven - all that knob twiddling has to do something to a person's mind.

In the early 1960's Dr. R. A. Moog (recognize the name?) began developing and producing a line of electronic music synthesis equipment that revolutionized the field. The feature that made the Moog equipment such a quantum jump in ease of operation sounds almost ridiculously simple, but its implications are so far reaching that it must be stressed; THE KEY PARAMETERS OF THE PROCESSING ELEMENTS ARE A FUNCTION OF THE SUM OF SEVERAL CONTROL VOLTAGES RATHER THAN THE POSITION OF A KNOB.

As an example of the operating ease of voltage control let's see what it does for a relatively simple processing element, an amplifier. As we shall see a little later, one of the things that contributes most to the way an instrument sounds is the manner in which its sound builds up and dies away. When using the classical tape splicing technique these characteristics have to be duplicated manually for every note by turning the volume control of the amplifier. Even though the Mark II allowed for automatic control of the amplifiers, information still had to be punched into its programming tape for each individual note.

With voltage control the job of setting the correct time varying amplifier gain can be turned over to an automatic electronic function generator circuit that produces a repeatable, pre-set voltage waveform each time a key is pressed. This voltage is then used to control the amplifier. The musician sets the function generator to reproduce the characteristics of some real or imagined instrument and the electronics will produce that characteristic for each note he plays. If he desires a totally different sound it's simply a matter of re-setting a couple of knobs. Summing the control voltages allows the performer to produce more than one effect from a single processing module. If, in the above example, the operator decides to add a low frequency amplitude modulation (tremolo) to the sound, he needs only to sum a second voltage that is changing at the rate of the desired tremolo into one of the remaining amplifier control inputs. As the control voltage varies up and down so does the gain of the amplifier and the volume of the sound.

THE SOUND OF MUSIC

Anyone can make weird noises on a synthesizer simply by randomly making connections and pushing buttons. It's even fun for the first hour or so, until you begin to think of specific sounds you want to make and can't. If we're going to learn to use a synthesizer rather than just play with it it's important that we understand what sound is and what makes one sound different from another.

If your knowledge makes the following discussion seem trite, read on anyway. We have to start somewhere and if nothing else you can probably find something to disagree with.

Sound travels as waves, waves of pressure in the air. A vibrating string displaces the air around it and the air molecules that the string moves in turn bump into and move other molecules. All the things that these sound waves can bump into and be reflected off of and the effect that this has on the original wave are beyond the scope of our discussion. The only thing relevant to the subject at hand is that if a man is present the pressure of the waves will finally cause a deflection of his eardrum which in turn will vibrate three small bones inside his ear which will in turn cause a disturbance in a fluid medium which in turn excites the auditory nerves which in turn causes the man to say " Hey, listen to that_____." Whether he fills in the blank with "noise or "music" is personal preference.

The thing that vibrates to produce the sound doesn't have to be a string. It can be a synthetic or organic membrane as in a drum, a vibrating reed as in the wind instruments or the lips of the musician as in the brass instruments. Most important to us, it can also be the cone of a loudspeaker.

When a recording of a musical instrument is made a microphone converts the air pressure waves into exactly analogous electrical voltage waves. If you were to graph the vibrations of the air and the "vibrations" of the voltage side by side they would be identical except that one would be measured in volts and the other in dynes per square centimeter - or something. When these voltage variations are re-played through an amplifier and loudspeaker they are converted from electrical back into sound energy. If all the links in the chain have been faithful in their recording and reproducing functions the pressure waves generated by the loudspeaker will be exactly the same as those originally generated by the musical instrument and the two will be indistinguishable.

Since the thing that an amplifier and loudspeaker works with is not really sound but an electrical analog of sound; and since it is possible to electronically generate any imaginable voltage waveform (difficult in some cases, but possible), it seems only logical that at some point sounds should be generated not by physical musical instruments but by synthesizing their electronic analog and then converting that to sound.

PITCH, DYNAMICS, TIMBRE

There are really only three characteristics that determine what a musical instrument will sound like; pitch, dynamics and timbre. Of the three, pitch probably requires the least explanation.

Pitch and frequency are two words from two different technologies that describe the same thing. When an engineer or technician speaks of 261 Hz, they mean that the thing they are referring to is vibrating 261 times per second. When a musician mentions middle C he is also talking about something that is vibrating 261 times per second. If the musician is dealing with conventional instruments he is probably talking about a string or reed but if he is working with an organ or synthesizer he is likely referring to the same thing that the technologists were talking about, the frequency of the changes of an electrical waveform.

The human ear is more sensitive to changes in pitch than any other musical parameter. The intensity of a sound has to be cut significantly before a listener experiences any decrease in loudness but a skilled musician can tell when a musical semi-tone deviates by as little as 3% of the interval between that note and the next higher tone.

Dynamics is a broad term that refers to the time varying intensity characteristics of the sound; how fast it builds up and how fast it dies away.

The length of time required for a sound to build up to its greatest intensity is called attack time and this one parameter conveys more information about the way an instrument is played than any other. If the attack time is very short the instrument will be in the percussion family where the vibrating member is immediately excited to its maximum amplitude by the deforming action of being plucked or struck with a hammer or mallet. If the attack is relatively slow then the instrument is probably in the reed or bowed string groups where the action of the exciting force - the wind or bow of the performer - takes a short time to fully excite the vibrating element.

If you forget about the talent factor for a moment the primary purpose of the musician in playing most instruments is to serve as an energy source. The performer pumps energy into the system (instrument) and the system dissipates it in some way, usually as either sound or heat. I know what you're probably thinking. Heat? Yes, heat; if you were able to accurately measure the temperature of a drum head you'd find that it gets hotter as you pound on it. The energy that is converted to heat can be thought of as being lost since it does not contribute to the primary object of producing sound.

Very interesting, right? But what has this to do with the sound of a bassoon. Just this, another important characteristic of an instrument is its release time. That is, how fast the sound dies away. Release time is directly related to how much of the energy goes into heat and how much into sound. A vibrating string, for instance, is as close to lossless as you can get and its release time is very long. The stretched membrane of a drumhead on the other hand is very lossy and as a result the release time of drums is very short.

Reed instruments have a short release time because the reeds are relatively lossy and don't continue to vibrate for very long after the musician stops adding energy. Brass instruments have the shortest release time because the performer can force his lips to stop vibrating and the column of air in the instrument is very lossy.

Sustain time is the interval in between attack and release, the steady state response of the instrument. As is obvious, percussion instruments have zero sustain time - as soon as the attack is finished there is no more energy input so it's downhill the rest of the way. Instruments that have some continuous energy input from the performer, in the form of bowing, blowing or even pedaling in the case case of some organs, can sustain as long as the energy holds out.

Though attack, sustain and release are the primary phenomena of dynamics there is one other condition that is common. When a percussion instrument is struck very hard the vibrating member will deform beyond the point at which a smooth release is possible, in effect more energy is put into the system than it can handle, with a resulting overload. Under these conditions the system (string, membrane or whatever) will rapidly get rid of the excess energy. With the "overload" dissipated the vibrating element will continue to dissipate the remaining energy in a normal fashion. The result is an initial rapid attack immediately followed by a decay time which is then followed by a normal release. In a natural instrument it would be all but impossible for the decay time to be followed by a sustain interval but with a synthesizer this is simple.

We can graphicly illustrate the conditions discussed by plotting the overall intensity of the sound versus time as shown in figure 1. Since these graphs are drawn to show the peak amplitude of the sound at any given time and therefore "contain" the sound they are often referred to as envelopes.

It is pretty obvious that as important as dynamics is, it doesn't account for all the differences between the sounds of instruments. For instance, the trumpet and french horn are both brass instruments with approximately the same attack, sustain and release characteristics. They even overlap as far as pitch range is concerned but there would be little danger of mistaking the blaring, brassy sound of the trumpet for the mellow, muted tones of the french horn. These differences come about

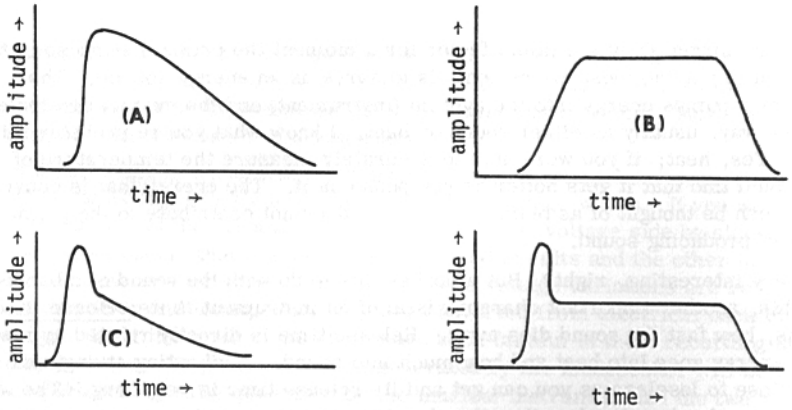


FIGURE 1. Amplitude envelopes for (A) Percussion (B) reeds (C) attack-decay-release (D) attack-decay-sustain-release

because no musical instrument produces a tone that is composed exclusively of a single frequency. Each note is composed of a number of different frequencies, and the number and amplitude of the various components are what gives each instrument its distinctive timbre.

The concept that a single musical pitch can be made up of more than one frequency can be confusing and needs further attention. The sine wave shown in figure 2 is the basic building block of any imaginable accoustic or electrical wave. It is the only waveform that is composed entirely of a single frequency and, more importantly, any waveform can be built up using nothing but sine waves.

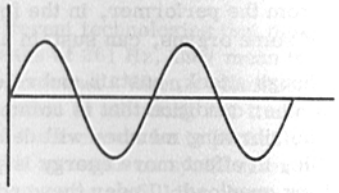


FIGURE 2. Sine wave

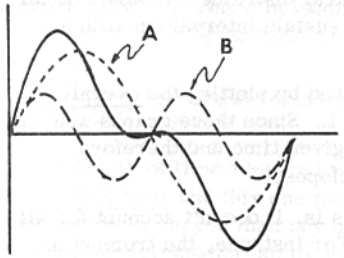


FIGURE 3. Fundamental and 2nd. harmonic

To illustrate this look at figure 3. Here we have two sine waves drawn in dotted lines which are labeled "A" and "B". As you can see from the drawing, waveform "B" goes through two cycles in the time that it takes waveform "A" to complete a single cycle. Waveform "B" is therefore twice the frequency of "A" and is said to be the second harmonic of the fundamental frequency "A". If we draw another wave that was three times the frequency of "A" it would be the third harmonic, four times would be the fourth harmonic, five times the fifth and so on.

If at every point in time we sum together the amplitudes of waveforms A and B the result is the waveform shown by the solid line. Note that while the new wave is shaped differently than either A or B it has the same frequency (and consequently pitch) as the fundamental frequency A. If third, fourth, fifth and higher order harmonics were added into this wave the result would continue to change shape but the frequency would remain the same.

It is not necessary that every harmonic of a fundamental frequency be included in a wave and indeed the most musically interesting sounds have certain harmonics deleted. The square wave shown in figure 4 is a good example. It is difficult to imagine that the sharp-edged wave illustrated could be built up from smoothly changing sine waves but it can as shown in the progression of diagrams figure 5 (a) through (c). In (a) fundamental frequency is added to its third harmonic producing the waveform shown by the solid line. In (b) the fifth harmonic has been added to the result of (a) to produce the new solid waveform and in (c) the seventh harmonic has been added to all the rest. You can see that the trend as higher order harmonics are added is to steepen the sides of the square and flatten and reduce the ripple in the top. When enough harmonics have been added the result will be a square wave. Notice in particular that not all harmonics are added together for a square wave, only the odd harmonics (3rd, 5th, 7th, etc.) are included.

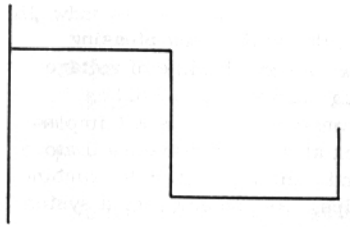


FIGURE 4. Square wave

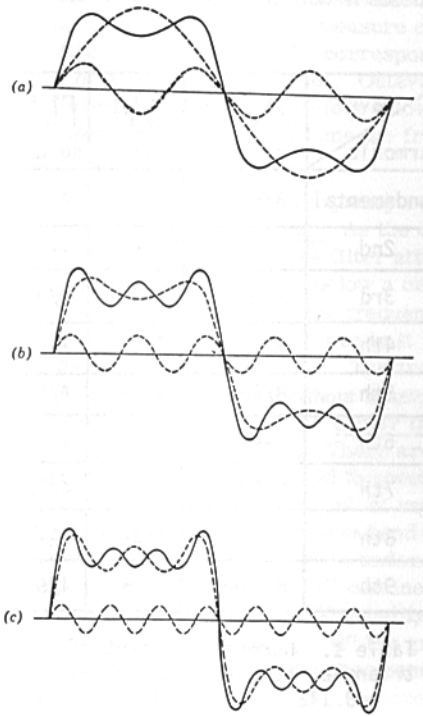


figure 5.

MAKING WAVES

Now that we have a pretty good idea of why instruments sound the way they do we can looking at ways of duplicating these sounds using electronic circuits.

The first method of electronically producing a desired waveform is called additive synthesis and the technique should be obvious from the discussion of harmonic structure. Several oscillators provide a source of sine waves of various harmonically related frequencies and combinations of the outputs are summed together to build up the desired waveform. By changing the amplitudes of the sine waves practicaly any waveform can be easily produced. One of the problems with this system is keeping all of the oscillators tuned so that they are multiples of one

another. Most electronic organs that use additive synthesis systems get around this problem by using one oscillator for the highest frequency component desired and then producing the other frequencies using a chain of frequency dividers.

The technique used in synthesizers is called subtractive synthesis and can be thought of as just the opposite of additive synthesis. Rather than summing together the frequencies you do want, you start off with a source that is already rich in harmonics and then remove the ones you don't want. This may seem a rather strange way to get from here to there but there is an excellent biological precedent for subtractive synthesis, the most versatile musical instrument of all - the human voice.

There are other reasons for using subtractive synthesis than just pleasing mother nature. If we are going to be consistent in our design of a line of voltage controlled equipment, then everything should be voltage controlled, including the oscillators. Designing a voltage controlled sine wave oscillator is not impossible but it is difficult. Then there is the need to lock all the sine wave oscillators to precise multiples of the fundamental frequency, and a mixing system to combine all the harmonics in the proper amplitude relationships. As you can see, a system of this type would be quite cumbersome and tedious to operate.

From a technological standpoint, it is much easier to electronically generate a complex ramp or square wave than a sine wave. And with the many recent refinements in filter circuits, subtractive synthesis is definitely the better route to take.

Since synthesizers operate with harmonic rich waveforms as their primary signal source there is no need to start out with a sine wave at all. The VCO's supplied with most synthesizers provide a variety of waveforms each of which provides different harmonic structures. Common practice is to use a relaxation oscillator to generate a voltage ramp which is then converted to triangle and pulse waves using simple shaping circuits. In some cases the triangle will also be shaped into a sine wave. These waveforms and their harmonic contents are listed in table 1.

In order to use subtractive synthesis we need some means of getting rid of the harmonics we don't want and to do this we use filters. A filter is quite simply an electronic gadget that eliminates a single frequency or group of frequencies.

wave / harmonic	triangle	ramp	square
fundamental	$8/\pi^2$	$2/\pi$	$4/\pi$
2nd	---	$1/\pi$	---
3rd	$8/9\pi^2$	$2/3\pi$	$4/3\pi$
4th	---	$1/2\pi$	---
5th	$8/25\pi^2$	$2/5\pi$	$4/5\pi$
6th	---	$1/3\pi$	---
7th	$8/49\pi^2$	$2/7\pi$	$4/7\pi$
8th	---	$1/4\pi$	---
9th	$8/81\pi^2$	$2/9\pi$	$4/9\pi$

Table 1. Harmonic content of triangle, ramp and square wave
π=3.142 π^2=9.872

Figure 6 shows diagramatically a representation of the frequency response of a low pass filter. This drawing shows that as the frequency of the signal being fed to the input of the filter increases the amplitude of the filter's output falls off. Note that the filter does not change the frequency of the input signal, only the amplitude. If the input is a complex waveform the filter will of course change the signal's shape as it attenuates the higher frequency components but that is, after all, what we're after.

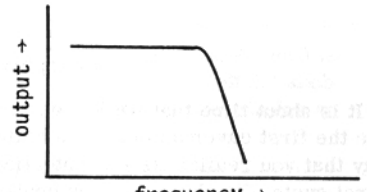


FIGURE 6. Low-pass filter

Figure 7 shows the frequency response of a high pass filter. In this case the amplitude of the output falls off as the input frequency decreases.

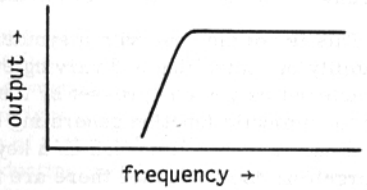


FIGURE 7. High-pass filter

Notice that in both of these filters the response curve is flat either up to or beyond some definite frequency. This is the frequency at which the filter begins to "take hold" and is designated the cutoff frequency or f_c. One other important parameter associated with low pass and high pass filters is the roll off rate, ordinarily measured in units of db/octave. This sounds complicated but it's really not. A decibel (db) is a measure of electrical level and when you're talking about voltage, a change of 6 db. corresponds to halfing (if - 6 db.) or doubling if (+ 6 db.) the original reference level. Octaves are of course frequencies that are double some reference frequency; therefore a low pass filter that "rolls off" at 6 db./octave simply means that every time the frequency is doubled the output of the filter falls by 1/2.

Figure 8 shows the frequency response of a band pass filter. As the diagram implies, a band pass filter attenuates all frequencies above and below a certain frequency while allowing the frequency of interest (or frequencies close to it) to pass without being effected. The frequency that is allowed to pass without attenuation is quite logically called the center frequency and is also designated f_c. There are parameters that can be used to specify how well the filter does its job of rejecting frequencies outside of its pass band but none of them are very easy to understand and for our purposes we will confine ourselves to speaking of the "Q" (quality) of the filter. The higher the "Q" the greater the frequencies outside the pass band will be attenuated.

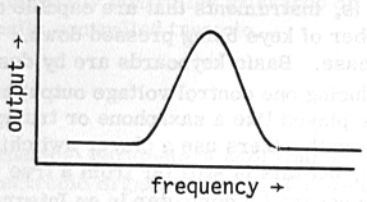


FIGURE 8. Band-pass filter

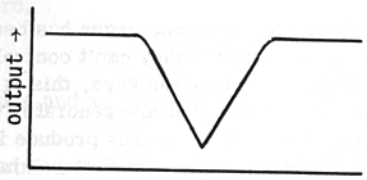


FIGURE 9. Notch filter

As the frequency response curve of figure 9 shows, you can think of a notch filter as being the opposite of a band pass filter. Instead of allowing frequencies around the center frequency through, the notch filter blocks these and allows all others to pass.

CONTROLLERS

It is about time that we looked at a problem that has plagued instrument makers since the first caveman beat on a hollow log - how to control the instrument in such a way that you realize its full potential. With most conventional instruments the control system is obvious. You control some of the elements of the dynamics by how hard you blow, pick, or strike the instrument and you control the pitch by the positions of your hands and/or lips. Timbre is in most cases a quality of the instrument and is therefore beyond the control of the performer.

This is not the case with a synthesizer; you have at least the theoretical capability of controlling and varying every characteristic of the sound. Some characteristics you can pre-set by the position of a knob and some you can turn over to automatic function generating equipment. Some parameters are varied with a manual controller such as a keyboard and some, unfortunately, you wind up forgetting about because there are no more controllers available.

Before examining some of the types of controllers that are available, make sure that you have firmly implanted in your mind that a controller for a synthesizer does only one thing; it provides a voltage proportional to some parameter that is physically changed by the performer. While in most cases the voltage produced by the controller will subsequently be used to set the pitch of a VCO, this will not always be the case. Depending on the sound being produced, the controller may also be used to set the center frequency of a band-pass filter, roll off rate of a low pass filter or any number of other things.

KEYBOARDS

When used to control a piano, a keyboard is one of the greatest inventions of all time. When used with a synthesizer it is at best a compromise.

Musicians are used to keyboards being connected to polytonic instruments; that is, instruments that are capable of playing as many notes at one time as the number of keys being pressed down. With most basic synthesizers, this is not the case. Basic keyboards are by design a monotonic controller capable of producing one control voltage output at a time. This means the synthesizer is to be played like a saxaphone or trumpet - one note at a time. Some intermediate line synthesizers use a clever switching arrangement to produce two notes at a time, but this is still far from a true polytonic system. Development of micro-processors (a computer in an Integrated Circuit) will be the technological step which will finally allow development of fully polytonic keyboards - playing more than one note at a time, and having individual voicing for each additional note being played.

Since the electronic organ has become commonplace, performers have gotten used to the idea that they can't control the dynamics of their instrument by varying the striking force on the keys, this is also true of most synthesizers. Other than triggering signals that are generated when any key is pressed, the only control voltage that most keyboards produce is proportional to the location of the key being activated. One manufacturer has a keyboard that is an exception to this rule; in addition to the standard control voltage it also generates two voltages proportional to the velocity of the key as it is pressed down and the final pressure on the key as it is held down. This is a significant improvement since it allows the performer to directly influence three musical parameters by pressing a single key.

Strangely enough, the original objection to a keyboard that was mentioned in the first part of this booklet (unavailability of pitches between semi-tones) is not a great problem on a synthesizer. Most keyboards provide a "pitch" knob that allows some variation in tuning of the instrument and many provide for an automatic, variable rate glissando.

In spite of its drawbacks, the standard keyboard has one big thing in its favor - familiarity. It is similar to a thing that the musician already knows how to use and re-training time is therefore reduced.

LINEAR CONTROLLERS

These are electrically and mechanically the simplest of all controllers. Most consist of a long strip of electrically resistive material with a voltage applied to each end. The potential difference between the two ends distributes evenly along the length of the strip so that the voltage between any point and electrical ground is proportional to the position of that point on the strip. When the performer presses on the controller, a conducting metal band makes contact with the resistance element and picks off the voltage present at the point of contact.

Linear controllers are generally not intended as substitutes for keyboards for a number of reasons. First, it is technically difficult to automatically produce a trigger pulse whenever the controller is pressed. This function has to be performed manually with a seperate switch that must be closed for each note or run that is to be played. Secondly, using a linear controller for pitch is like playing a fretless instrument such as a violin, it requires considerable experience to know what pitch is going to be produced at a given location.

These devices come into their own when used in conjunction with a keyboard. In this application they can provide an auxillary control for some parameter other than pitch, like manually sweeping a filter or controlling the amount of noise mixed into a sound. The control voltages produced by this unit can also be summed into one of the VCO control inputs to produce a manually controlled glissando or vibrato or can be used with a VCA to give manually controlled tremolo.

FOOT PEDALS

Foot pedals allow you to control additional musical parameters with your feet. They are similar to the expression pedals on electronic organs except that instead of controlling only the volume they can be used to control filters, oscillators or amplifiers.

Anything you can say about linear controllers applied to foot pedals, they're intended to be used in conjunction with a keyboard.

JOY STICKS

These are the wackiest controllers imaginable and as you would expect are similar to the joy sticks used in airplanes.

The biggest thing going for this type of controller is that it offers the possibility of directly controlling four musical parameters simultaneously. One parameter could be controlled by moving the stick forward and backward, another by moving the stick from side to side, a third control voltage could be generated proportional to vertical motions (along the long axis of the stick) and a fourth porportional to

the rotation of the handle. If you like you could even put a switch on top to control such vital functions as self-destruct.

A joy stick seems like a valid concept but anyone that could use one properly probably wouldn't be able to communicate with earth people.

ENVELOPE GENERATORS

Envelope generators are automatic controllers that electronically generate a time varying voltage as pre-set by the positions of knobs or sliders. An envelope generator ordinarily responds to a trigger pulse by generating an electrical waveform that rises to some pre-set value in a pre-set time, sustains that level as long as the trigger pulse is present (or for a pre-set time in some cases) and then falling back to zero in a pre-set time. Some envelope generators are capable of producing the attack, decay, sustain, release type functions discussed earlier.

The output of the envelope generator can be used in the same ways any other control voltage source should, but these items find their most common application in controlling dynamics and time varying timbral qualities of a sound.

A low frequency oscillator can also serve as a control voltage source to provide cyclicly varying voltages for vibrato, tremolo or filter sweeping.

SEQUENCERS

Sequencers fall under the category of control devices much like keyboards. The main point of a sequencer is that it is programmed ahead of time to provide a specific pattern of control voltages. On command, the series of voltages appears at the output, and this pattern can be repeated many times with high consistency. Standard analog sequencers are programmed via potentiometers to provide the desired voltage for each stage. After programming, the module is turned on either manually or from remote trigger signals, and the internal clock steps through the sequence to "read" the programmed voltages. Digital technology has aided in the development of digital sequencers which differ in one main area. Rather than manually programming the voltage for each stage, an input is provided to accept control voltages from a keyboard or other control device. As the keyboard is played, the various control voltage steps are translated into a digital word and stored in a memory system. On command, the sequencer reads its output information from the memory rather than from a string of potentiometers. These devices are still costly, but as computer technology increases the cost will continue to drop.

Although sequencers are usually thought of as being used to control an oscillator (repeating bass lines, high speed arpeggios etc.) many interesting effects can be obtained by using them as controllers for filters, VCA's or any other modules requiring a control voltage.

DIGITAL COMPUTERS

As short as two years ago, thoughts of digital computers being interfaced with an electronic music synthesizer were reserved for well funded college music departments or government subsidized artistic research programs. Since then, computers have grown smaller, more inexpensive, and basically more accessible. Electronic hobbyists are delving into these machines as they did with Ham radio ten years ago. Realizing a computerized music system is

practical today. Initially, we will see computers being used to control the analog sound generating and processing modules of the synthesizer. Then, as prices decrease, fully computerized synthesizers will be seen which will generate any waveform, changing waveform, amplitude, and other parameters as you play - depending on how the computer is programmed. Eventually, we may even see computer music systems which can "listen" to a sound and reproduce the same sound under operator command. Or how about a machine that monitors the operators brain waves and vital signs, and composes music specifically to fit his moods/thoughts. The list is endless.

THE EQUALLY TEMPERED SCALE

This is as good a place as any to bring up the subject of the equally tempered or chromatic scale. As anyone who is reading this knows there are 12 semi-tones in each octave of the chromatic scale, 7 naturals labeled A through G and 5 accidentals that are designated as either sharps or flats of the naturals. With two exceptions the sharp of one note is identical to the flat of the next highest note, the exception being that there are no accidentals between B and C, or E and F so that B# is the same as C and F^b is the same as E.

For each octave increase in the musical scale the frequency of the note doubles so that since middle C corresponds to 261.6 cycles per second the next C above middle C is 523.2 cycles per second.

Somewhere back in antiquity (around the time of J. S. Bach) some genius decided that since there are 12 semi-tones to the ocatve and each octave doubles the frequency, each note should be related to the note directly below it in the scale by a factor of the twelfth root of two. Just in case you're not used to working out the twelfth root of numbers in your head, this translates to 1.059 times times the frequency of the note directly below it. The significance of this is that as pitch increases, the difference between adjacent notes in the scale also increases.

All this may seem like academic trivia until you realize one point. All voltage controlled oscillator designs produce a device whose output frequency is directly proportional to the control voltage, identical control voltage changes produce identical frequency changes.

An example will most readily demonstrate the significance of these facts. Suppose that we have a keyboard that produces a control voltage of .625 volts when its lowest C is pressed. The voltage corresponding to the next C up is quite logically 1.25 volts but don't fall into the trap of thinking that the voltage corresponding to the third C is 1.25 plus .625 or 1.875 volts because it's not, it should be twice the voltage required for the second C or 2 X 1.25 = 2.5 volts.

Many synthesizer designers use an electronic conversion device to get around this difficulty. This device converts a linear controller output voltage (1v. for the first C, 2v. for the second, 3v. for the third C, etc.) to the octavely related voltage required by the VCO.

Unfortunately, the exponential converter circuits (as these devices are known) are not only expensive but also quite often they tend to drift so that even for a fixed input voltage the output voltage (and of course pitch of the VCO) wanders from one value to another.